## CLAIMS

*No claims are currently being amended, added or canceled. All pending claims are reproduced below.*

1. (Previously Presented) A method comprising:

storing a plurality of independent sets of filter coefficients in a memory, wherein each set of filter coefficients defines a different polyphase filter function, wherein each of the different polyphase filter functions would result in at least some modifying of a signal if the signal were filtered in accordance with the polyphase filter function, and wherein each of the different polyphase filter functions would result in modifying of a signal in a different manner than the other polyphase filter functions;

selecting one of the sets of filter coefficients;

interpolating the selected set of filter coefficients to thereby produce interpolated selected filter coefficients; and

convolving the interpolated selected filter coefficients with an input signal to produce a filtered output signal that differs from the input signal regardless of which one of the sets of filter coefficients is selected.

2. (Previously Presented) The method of claim 1, wherein the input signal comprises an audio signal, wherein the input signal is convolved with the interpolated filter coefficients in a sample rate converter of a digital pulse width modulation (PWM) audio amplifier.

3.-4. (Canceled)

5. (Previously Presented) The method of claim 1, wherein selecting one of the sets of filter coefficients comprises reading a value stored in a filter selection register and selecting one of the sets of filter coefficients based upon the value.

6. (Previously Presented) The method of claim 5, further comprising changing the value in the filter selection register to a new value and selecting a new one of the sets of filter coefficients based upon the new value.

7. (Original) The method of claim 1, wherein the plurality of sets of filter coefficients are stored in a single memory.

8. (Previously Presented) The method of claim 1, wherein the selected set of filter coefficients are interpolated according to a cubic spline algorithm.

9. (Original) The method of claim 1, wherein each of the plurality of sets of filter coefficients comprise polyphase filter coefficients.

10. (Previously Presented) A system comprising:

a coefficient interpolator; and

a memory coupled to the coefficient interpolator;

wherein the memory is configured to store multiple independent sets of filter coefficients, wherein each set of filter coefficients defines a different polyphase filter function, wherein each of the different polyphase filter functions would result in at least some modifying of a signal if the signal were filtered in accordance with the polyphase filter function, and wherein each of the different polyphase filter functions would result in modifying of a signal in a different manner than the other polyphase filter functions; and

wherein the coefficient interpolator is configured to interpolate a selected one of the sets of filter coefficients to thereby produce interpolated selected filter coefficients.

11. (Previously Presented) The system of claim 10, further comprising a convolution engine coupled to the coefficient interpolator and configured to convolve an input signal with interpolated coefficients corresponding to the selected one of the sets of filter

coefficients to produce an output signal that differs from the input signal regardless of which one of the sets of filter coefficients is selected.

12. (Previously Presented) The system of claim 11, wherein the convolution engine is configured to convolve an audio input signal with the interpolated coefficients to produce an output audio signal, wherein the convolution engine is implemented in a sample rate converter of a pulse width modulation (PWM) amplifier.

13.-14 (Canceled)

15. (Original) The system of claim 10, further comprising a filter selection register configured to store a filter selection value, wherein the coefficient interpolator is configured to interpolate a set of filter coefficients indicated by the filter selection value in the filter selection register.

16. (Original) The system of claim 15, wherein the filter selection register is configured to allow modification of the filter selection value.

17-18. (Canceled)

19. (Original) The system of claim 10, wherein the memory comprises a single memory module configured to store the multiple sets of filter coefficients.

20. (Previously Presented) The system of claim 19, wherein each of the multiple independent sets of filter coefficients comprise polyphase filter coefficients.

21. (Original) The system of claim 10, wherein the coefficient interpolator is configured to interpolate the selected set of filter coefficients according to a cubic spline algorithm.

22. (Previously Presented) A method comprising:

storing a plurality of independent sets of filter coefficients in a memory, wherein each set of filter coefficients defines a different polyphase filter function, wherein each of the different polyphase filter functions would result in at least some modifying of a signal if the signal were filtered in accordance with the polyphase filter function, and wherein each of the different polyphase filter functions would result in modifying of a signal in a different manner than the other polyphase filter functions;

selecting only one of the sets of filter coefficients;

interpolating the selected set of filter coefficients to thereby produce interpolated selected filter coefficients; and

convolving the interpolated selected filter coefficients with an input signal to produce a filtered output signal that differs from the input signal regardless of which one of the sets of filter coefficients is the only one selected.

23. (Previously Presented) The method of claim 22, further comprising performing the method in a sample rate converter of a digital PWM amplifier, wherein the input signal comprises an audio signal.

24. (Previously Presented) The method of claim 1, wherein the plurality of sets of filter coefficients are stored in the memory prior to receiving the input signal, and wherein the filter function defined by each set of filter coefficients corrects distortion in the output signal.

25. (Previously Presented) The system of claim 10, wherein the memory is configured to store the multiple sets of filter coefficients prior to receiving an input signal, and wherein the filter function defined by each set of filter coefficients corrects distortion in an output signal produced by convolving the input signal with interpolated coefficients based on the corresponding set of filter coefficients.

26. (Previously Presented) The method of claim 22, wherein the plurality of sets of filter coefficients are stored in the memory prior to receiving the input signal, and wherein the filter function defined by each set of filter coefficients corrects distortion in the output signal.

27. (Previously Presented) The method of claim 1, wherein the output signal, resulting from the convolving step, is dependent on which one of the independent sets of filter coefficients is selected, such that for the same input signal a different output signal would be produced if a different one of the independent sets of filter coefficients were selected.

28. (Previously Presented) The system of claim 11, wherein the output signal, produced by the convolution engine, is dependent on which one of the independent sets of filter coefficients is selected, such that for the same input signal a different output signal would be produced if a different one of the independent sets of filter coefficients were selected.

29. (Previously Presented) The method of claim 22, wherein the output signal, resulting from the convolving step, is dependent on which one of the independent sets of filter coefficients is the only one selected, such that for the same input signal a different output signal would be produced if a different one of the independent sets of filter coefficients were the only one selected.